Date: Thu, 5 Aug 93 04:30:34 PDT
From: Ham-Homebrew Mailing List and Newsgroup <ham-homebrew@ucsd.edu>
Errors-To: Ham-Homebrew-Errors@UCSD.Edu
Reply-To: Ham-Homebrew@UCSD.Edu
Precedence: Bulk
Subject: Ham-Homebrew Digest V93 #2
To: Ham-Homebrew

Ham-Homebrew Digest   Thu, 5 Aug 93   Volume 93 : Issue   2

Today's Topics:

Date: 4 Aug 93 13:37:08 GMT
From: newsserv.cs.sunysb.edu!rick@nyu.arpa
Subject: Acquiring Crystal Filters
To: ham-homebrew@ucsd.edu

Bill Kirkland (kirkland@nimios.eng.mcmaster.ca) wrote:
: Does anyone out know of suppliers for crystal filters (i.e. single quantities)
: for 9 - 10 MHz IF and 40 MHz IF or higher?

: Thanks Bill Kirkland

Digikey sells 10.7 mHz crystal filters (2/4 pole) - see the July-Aug '93
catalog pg 119. I spent 2 weeks digging around for 45 mHz crystal
filters a few months ago: bottom line is that, outside of Japan, they are
basically either unavailable in < 100 quantity, or "made to order" and
very expensive. If loss is not a factor, and price is no object, you can
get SAW filters from Sawtek or Andersen pretty easily..

: reply to kirkland@comres3.Eng.McMaster.CA

Rick Spanbauer, WB2CFV
SUNY/Stony Brook

Date: Thu, 5 Aug 1993 08:28:02 GMT
From: mcsun!sun4nl!ruuinf!ruunfs.fys.ruu.nl!vreeburg@uunet.uu.net
Subject: Looking for SSB-chip
To: ham-homebrew@ucsd.edu

Hi there,

I'm looking for a single chip which is able of single-sideband-modulation. Not that I want to build my own transmitter, but I want to do a frequency shift of an audio signal by applying the audio signal and a 100Hz 'carrier' signal to this SSB-chip. The output signal, as far as I know, will be: audio + 100Hz.

Could someone point me to any manufacturor of such a chip.

Thanks.

--
Jurriaan Vreeburg   email: vreeburg@fys.ruu.nl
Department of Physics and Astronomy   tel.: (+31)-(0)30-534566
P.O. Box 80.000
3508 TA Utrecht Holland

Date: 4 Aug 1993 13:08:19 GMT
From: pipex!sunic!news.funet.fi!uta.fi!vuokko!stkaba@uunet.uu.net
Subject: Prescaler 10/11 needed ...
To: ham-homebrew@ucsd.edu

Hi, I have tried to find either Motorola or Plessey 10/11 prescaler for under 200Mhz operation, but they seem to be hard to get at low quantities.

Now, does anybody have such one in the junkbox and willing to sell it ?.

73 kari

--

Kari Back   E-mail stkaba@uta.fi
Riistakatu 3   Tel. 358-31-534661
33520 Tampere, Finland

Date: 4 Aug 93 15:33:06 GMT
From: uplherc!wpsun4!mb@uunet.uu.net
Subject: PUFF?
To: ham-homebrew@ucsd.edu

beyer@athena.uto.dec.com (Peter Beyer) writes:
:
: In article <careyj.743980901@spot.Colorado.EDU>, careyj@spot.Colorado.EDU (CAREY JOSEPH M) writes:
:
: |>My guess is 2.0 is the last version we're gonna see. I think it
: |>is three years old now. Written by grad students at Cal-Tech, I think
: |>they all granulated and went on to new projects.
:
: That's too bad. I use PUFF from the very beginning and still, I'am very pleased
: with it, though as said before, it's missing some important aspects. Commercial
: packages like Touchstone and SuperCompact are expensive, out of reach for
: the every Ham. If the items I mentioned were included it would be real
: competitive.
:

I just tuned in to this newsgroup, and missed the beginning of this
thread. I've checked Archie unsuccessfully for PUFF. Can anyone tell
me where I can find it?

Thanks much!
Michael Bendio, WT7J (mb@titan.wordperfect.com)

Date: 4 Aug 93 06:35:15 GMT
From: psinntp!psinntp!arrl.org@RUTGERS.EDU
Subject: Single frequency receiver
To: ham-homebrew@ucsd.edu

In rec.radio.amateur.homebrew, gary@ke4zv.uucp (Gary Coffman) writes:
>In article <1925@arrl.org> zlau@arrl.org (Zack Lau) writes:
>>

>> Yes, amplifiers can certainly become oscillators. However, many
>>amateur stations generate a -10 dBm signal that becomes a +60 dBm signal,
>>all on the same frequency! Thus, it certainly is possible to get 70
>>dB of gain on one frequency. Since tuned radio frequency receivers use
>>two frequency bands (audio and RF), getting 140 dB of gain out of a TRF
>>is certainly feasible. By going to 3 frequency bands, (audio, IF, and
>>RF), you can have a receiver that has very little shielding--just a good
>>layout. This is quite important to a manufacturer who wants to make
>>millions of cheap receivers at a competitive price.
>
>Or amateurs with less than optimum construction skills. It's certainly
>easier to tame a system that starts with internally generated drive at
>-10 dbm than one that starts with external signals at -120 dbm for most
>constructors. Keeping the feedback leakage that low can be a challenge.

Not in my experience. Receivers are usually much easier to build than
transmitters that have a -10 dBm low level signal, despite the much
higher gain. The difficulty with transmitters is that you have to watch
the grounding much more carefully--high current flowing on a "ground" plane
can do nasty things to low level circuits. Receivers rarely have high
current circuits except at audio. Yes, some high performance receivers
have high bias currents, but these are static currents that really don't
disrupt other circuits. Perhaps this is the reason you see more published
CW than SSB transmitters--you can start off with +10 dBm on CW, which means
a 10 watt transmitter may have only 30 dB of gain. Published SSB
transmitters are actually quite rare--once every couple years is the norm.

>It's certainly not impossible, as TRF receivers of the past demonstrate.
>
>> Finally, there is a good reason for not using mixers if at all possible.
>>They are typically the weak link in the system--they generate distortion
>>products that are often objectionable. Mechanical filter and crystals
>>can also cause problems, particularly when hit with too much signal. Thus,
>>direct conversion and TRF systems often sound much better that more
>>conventional commercial approaches.

It might be interesting to do a DSP receiver with a TRF front end
when the technology gets cheap enough for your budget. It ought to be
possible to to an "intelligent" detector, one that can detect interference
on one sideband and switch to the other. The band limited signal
provided by the front end greatly eases the requirements of the A/D
converter greatly. One of these days I'll have to dust of my textbooks
on DSP/digital design. I actually have a photocopied book that is
completely legal--the visiting MIT professor ran off a few copies of a
computer vision textbook he was working on :-).

>
>They can indeed. I think that taming these beasts can be much more

>of a challenge than with a superhet, however. Many DC receivers,
>where all the gain is at audio frequencies, have stability problems,
>and hum and noise pickup problems, not to mention image problems.
>Still, some perform remarkably well. And old TRF receivers were

Or, you could use intelligent design. If you use the balanced output
of an NE602 to drive a balanced amplifier (properly wired up OP AMP),
the hum and noise problems go away. The image can certainly be a
problem, particulary if you have CW stations every 200 Hz. Of course,
this means that well over 250 stations are stuffed into 50 kHz. But,
when things aren't as busy, you can offset the receiver a little,
which makes the pitch of the desired station go one way and that of
the inteference the other. The single board transceiver in the ARRL
Handbook allows this.

>incredibly touchy to tune, easily breaking into oscillation with
>the wrong combination of settings. Even the wind blowing the antenna
>would often trigger them into oscillation. A TRF built for a single
>frequency might perform well if constructed well, but it wouldn't be
>my first choice due to the numerous chances for problems.

>
>I've listened to old TRF radios and been amazed at their clarity compared
>to superhets of the same vintage, but they don't have AGC, and the tubes
>of the day were more non-linear than any of the circuits used in modern
>superhets, and the shape factor of their LC networks doesn't compare to
>that of modern crystal or mechanical filters. All but the latter might be
>alleivated in a modern design. Perhaps even the shape factor could be
>improved with proper stagger tuning on a fixed frequency receiver. It

Yes, even shape factor can be improved to match that of a crystal
factor, though probably not if you are trying to use the concept of
stagger tuning when doing the design.

I'd get a copy of Zverev's Handbook of Filter Synthesis, which has
tables for designing fancy filters like approximate Gaussian filters.
There probably is some shareware program that will do the design,
but I'm not aware of it.

The disadvantage of LC filters is that the Q of inductors is much
lower, so that you have to do the filtering at much lower frequencies
to get as narrow a bandwidth. On the other hand, crystals are series
resonant circuits with a stray parallel capacitance, which isn't
exactly the easiest component to deal with. If the Q is high enough,
you can get any shape factor you can get with crystals, provided
you use enough resonant circuits.

Zack Lau KH6CP/1

Internet: zlau@arrl.org "Working" on 24 GHz SSB/CW gear
Operating Interests: 10 GHz CW/SSB/FM
US Mail: c/o ARRL Lab 80/40/20 CW
225 Main Street Station capability: QRP, 1.8 MHz to 10 GHz
Newington CT 06111 modes: CW/SSB/FM/packet
amtor/baudot
Phone (if you really have to): 203-666-1541

>might be an interesting challenge to build a modern TRF, but if I wanted
>a practical receiver, I'd likely stay with a superhet.

Date: 5 Aug 93 01:08:47 GMT
From: ogicse!emory!kd4nc!ke4zv!gary@network.ucsd.edu
Subject: Single frequency receiver
To: ham-homebrew@ucsd.edu

In article <1954@arrl.org> zlau@arrl.org (Zack Lau) writes:
>
>Not in my experience. Receivers are usually much easier to build than
>transmitters that have a -10 dBm low level signal, despite the much
>higher gain. The difficulty with transmitters is that you have to watch
>the grounding much more carefully--high current flowing on a "ground" plane
>can do nasty things to low level circuits. Receivers rarely have high
>current circuits except at audio. Yes, some high performance receivers
>have high bias currents, but these are static currents that really don't
>disrupt other circuits. Perhaps this is the reason you see more published
>CW than SSB transmitters--you can start off with +10 dBm on CW, which means
>a 10 watt transmitter may have only 30 dB of gain. Published SSB
>transmitters are actually quite rare--once every couple years is the norm.

Hmmm. Good point. Maybe that's why I like modular construction, it
isolates those high ground currents. Of course it can cause other
problems. :-(

>It might be interesting to do a DSP receiver with a TRF front end
>when the technology gets cheap enough for your budget. It ought to be
>possible to to an "intelligent" detector, one that can detect interference
>on one sideband and switch to the other. The band limited signal
>provided by the front end greatly eases the requirements of the A/D
>converter greatly. One of these days I'll have to dust of my textbooks
>on DSP/digital design. I actually have a photocopied book that is
>completely legal--the visiting MIT professor ran off a few copies of a
>computer vision textbook he was working on :-).

I suspect we'll see DSP as IF replacement rather than as a TRF stage

follower because it pushes DSP frequency limits less and makes the software simpler. I'm looking forward to it, however, it does make building detectors with the keyboard possible.

>Yes, even shape factor can be improved to match that of a crystal
>factor, though probably not if you are trying to use the concept of
>stagger tuning when doing the design.

>The disadvantage of LC filters is that the Q of inductors is much
>lower, so that you have to do the filtering at much lower frequencies
>to get as narrow a bandwidth. On the other hand, crystals are series
>resonant circuits with a stray parallel capacitance, which isn't
>exactly the easiest component to deal with. If the Q is high enough,
>you can get any shape factor you can get with crystals, provided
>you use enough resonant circuits.

There's the rub, LC filters with narrow bandwidths and good shape are hard except at low frequencies. I assume by low you're probably talking 85 kHz or so like certain older commercial equipment used. That's not much use to a TRF design unless you're building a VLF recevier.

I suggested the expedient of stagger tuning because the insertion loss of high order LC filters can be quite high, the design is already requiring a lot of gain on frequency, and the problem is likely one of too narrow a peaked response rather than too broad. Stagger tuning is a venerable technique for broadening a very peaked response in a high gain amplifier.

Gary
--

```
Gary Coffman KE4ZV           |    You make it,     | gatech!wa4mei!ke4zv!gary
Destructive Testing Systems  |    we break it.     | uunet!rsiatl!ke4zv!gary
534 Shannon Way              |    Guaranteed!      | emory!kd4nc!ke4zv!gary
Lawrenceville, GA 30244      |                     |
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Date: 4 Aug 93 14:05:48 GMT
From: ogicse!uwm.edu!math.ohio-state.edu!cs.utexas.edu!not-for-mail@network.ucsd.edu
Subject: Source for Feeny's filter
To: ham-homebrew@ucsd.edu

This posting is an answer to Kevin Feeny's request re: spacial perception CW filter. It is a new posting as my mail reader has a problem where I can't continue a thread, or post directly. I'm posting via email to a host which (hopefully) will post this under my name.

The filter Feeny asks about is discribed in 1981 ARRL HANDBOOK pp 8_50-54.
ARRL did not mention the author or QST publication dates in this vintage
of handbook

The filter uses 6th order (two cascaded 3rd order Butterworths) filters,
A HP on the right ear, and a LP on the left. The article gives the 3 dB
points for the two 3rd order sections in cascade, as 583 Hz for the LP and
840 Hz for the HP. The responses cross at 700 Hz which is the -6dB point
for both filters.

I smelled a rat when I saw the two cascaded 3rd order sections, So I
calculated the transfer function of one of the LP sections. It has But-
terworth(well, within component tolerances, anyway) pole configuration,
with the break frequency at ~736 Hz. The reason for my suspicion was the
fact that two 3rd order sections in cascade can never give a Butterworth,
Chebyshev, elliptical, or otherwise optimized 6th order response. The
third order sections mandate the inclusion of two real poles, thus
severly rounding the "corner" of the response. I suspect that this is
why the curves cross at the -6dB point instead of the -3dB point which
would have been the obvious thing to do. At the -3dB point the filter
responses are still rather flat due to the choice of pole placement.

Heres how I will try to do it:
I will build true 6th order butterworth filters with all poles at
700 Hz. This will put the combined (HP+LP) response 3 dB above the skirts,
unless I blow the phasing which would give a deep notch there. (this is the
reason all _good_ loudspeaker crossover networks are odd-ordered) (OK golden
ears, when you flame, remember I said _good_ not expensive, which in the
fantasy land of audiophila rarely corilates)

Here is my $.02 WRT the phase vs. amplitude discussion in the other thread:
People who measure such things have pegged the bandwidth of the human
nervous system at around 10 Hz. This means that there is no physical
mechanism whereby we can directly sense the phase of an audio signal.

Yes, ears respond to sound well above that 10 Hz, not to mention the
frequency of light to whichour eyes are sensitive. In the case of the ear,
the sound produces a sypathetic vibration in one or more of the tuned
"reeds" in the cocula(sp?). The nerve cells respond to the _amplitude_ of
this vibration, and tell the brain how loud that sound is. The nervous
system hasn't the speed to tell the brain "now the pressure is rising,
now it's falling" as would be required to convey phase information.

This is not to say that phasing does not matter. In the case of stereo
loudspeakers, the phasing produces standing waves (OK, they only stand as
long as a given note lasts) which result in _amplitude_ differences at the
sides of your head. Thus we do _indirectly_ sense the relitive phasing of

two sources. Due to the less than perfect L-R channel isolation inside the skull, it is probable that some standing waves happen in there, which would allow some sensitivity to phasing while wearing headphones. As for the assertion that it is the phase difference betwixt the two ears which allows us to locate a single point source.. hogwash! Each ear has a directional pattern which has a peak about 45 degrees to the associated side. Thus when a sound arrives from one side it has a higher _amplitude_ on that side. The "horn antennas" (those flaps of skin on the side of my head) lose their directivity at low frequencys, allowing the sub-woofers to be placed wherever it's most convienient. Still not convinced? OK, a 10 Khz sound has a a wavelength of around 1 inch. Thus as the head is turned 90 degrees, the relitive phasing will change by roughly 720 degrees. Yet the direction to the sound can be accuratly sensed through the whole turn. I hope no ham is silly enough to suggest that it is possible to tell the difference between 90 deg phasing and 450 deg phasing: it just don't matter! Oh, if you conduct this test, do it outdoors or in an anechoic chamber: any reflective walls will make it nearly impossible to tell the direction, since the ears will pass through standing waves as the head is turned, resulting in wild variations in _amplitude_ .

To observe how much we rely on _amplitude_ differences, put on a pair of stereo 'phones, close your eyes, stand on one foot, and see how long you can keep your balance while your wife works the balance control back and forth.

The "delay and sum" schemes mimic the amplitude variations which occur when music from two sources is summed accusticly. The better ones reduce the amplitude of the delayed signals to mimic the directivity of the ear. It is the amplitude variations resulting from the summing process which produce the directional effects, not the phasing at all.

Sorry to run on like that, but the pseudo scientific hyperbole used to extract huge sums of money from self-styled "audiophiles" is one of my pet peaves. e.g. if an amplifier inverts, and you think it matters (it doesn't) then for crying out loud, reverse the damned speaker leads already!

So to get down from my high horse:

Kevin, If you can't find the '81 handbook then let me know. I'll be happy to copy the article, and send it to you for the cost of a SASE. If you got my earlier e-mail on this subject, note the minor changes here, as I was going from memory then, but have the handbook in front of me now.

-73-

Kevin KO0B (BAD in callbook) kevin@aquilagroup.com

-My employer shares my opinions..NOT!

End of Ham-Homebrew Digest V93 #2
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